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For: METHOD AND SYSTEM FOR ACOUSTIC SHOCK PROTECTION

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Attached please find the certified copy of the foreign application from which priority is claimed for this case:

Country	**: Canada**
Application Number	**: 2,424,093**
Filing Date	**: March 31, 2003**

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Specification and Drawings, as originally filed, with Application for Patent Serial No: **2,424,093**, on March 31, 2003, by **DSPFACTORY LTD.**, assignee of Todd Schneider, Robert Brennan and David Hermann, for "Method and Device for Acoustic Shock Protection".

Agent certificateur/Certifying Officer

March 26, 2004

Date

Canada

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ABSTRACT

An acoustic shock protection method and device are provided. A pattern analysis based approach is taken to an input signal to perform a feature extraction. A parameter space is identified, which is corresponding to the signal space of the input signal. A rule-based decision approach is taken to the parameter space to detect an acoustic event. The device may be advantageously implemented using a weighted overlap add approach to provide low group delay, high-fidelity and a high degree of protection from acoustic shock events.

Method and Device for Acoustic Shock Protection

FIELD OF INVENTION:

This invention relates to signal processing technology, and more particularly, to a method and device for acoustic shock protection.

BACKGROUND OF THE INVENTION:

Unwanted sounds, such as loud sounds or sounds that have a rapid relative increase in level, may be produced by telephone or radio systems, intentionally or unintentionally. Those sounds are typically experienced by a user through headphones or a headset.

Without protection against those sounds, the user may experience a phenomenon known as acoustic shock. Acoustic shock may result in permanent hearing loss, temporary hearing loss and tinnitus (constant ringing in the ears). Sufferers also report symptoms including extreme pain, vertigo and burning sensations. One of the main issues in the cause of acoustic shock syndrome, apart from the initial high-level sound, is the startle reflex action. This reaction can cause numerous muscles to activate to an unusual degree.

In order to prevent the user from experiencing acoustic shock, the following approaches are provided in telephone systems: 1- High-level using automatic gain control; 2- Adjustable notch filters to remove narrow band tones or "shrieks" when they are detected; 3- Clipping of high level signals using diodes or similar devices:

These approaches have also been combined. All of these approaches use techniques that are well known in the art and have been seen in other application areas (e.g. hearing aids).

The existing devices offer some protection. However, the processed output signal of those devices has reduced fidelity compared to an input signal. Typical distortions of the signal include "pumping" (unnecessary and audible gain adjustments of the gain that affect the speech signal) and "holes" (audio dropouts in

the processed signal caused by extreme gain adjustments), harmonic distortion as well as the accompanying intermodulation distortion that comes from poor gain control. More complex systems may also suffer from excessive input-output latency (i.e., group delay), which can adversely impact network, acoustic and line echo cancellers.

Currently there are specifications under development that provide guidelines and recommendations for the performance of acoustic shock systems. These include:

1- ITU-T Recommendation P.360 "Efficiency of devices from preventing the occurrence of acoustic pressure by telephone receivers";

2- UK standard BS6317 specified continuous signals only;

3- US standard, UL1950;

4- Telstra TT4;

5- EN60950 (see http://www.ktl.com/whatsnew/acousticshock.htm);

To meet these emerging performance requirements and deliver high-fidelity with low group delay, a new approach is needed. It is, therefore, desirable to provide a new method and device that can fully protect a user against the acoustic shock so as to meet the above guidelines and future guidelines that may emerge.

SUMMARY OF THE INVENTION:

It is an object of the invention to provide a novel method and device that obviates or mitigates at least one of the disadvantages of existing systems.

In accordance with an aspect of the present invention, there is provided a method and device for acoustic shock protection, in which a pattern-based detection approach and a rule-based approach for gain control (i.e., an expert system) are taken.

In accordance with a further aspect of the present invention, there is provided an acoustic shock protection method and device, in which a weighted overlap-add (WOLA) filterbank is used and a delay unit is used prior to WOLA analysis enabling predictive features to be fed around the WOLA filterbank analysis stage.

Here, the inherent delay of the WOLA analysis filterbank processing is advantageously used reducing the necessary amount of added delay maintaining a low overall group delay. The low group delay of the WOLA filterbank is a considerable advantage.

Other aspects and features of the present invention will be readily apparent to those skilled in the art from a review of the following detailed description of preferred embodiments in conjunction with the accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWINGS:

The invention will be further understood from the following description with reference to the drawings in which:

Figure 1 is a schematic diagram showing an acoustic shock protection system in accordance with an embodiment of the present invention;

Figure 2 is a schematic diagram showing the relationship between a signal space, analysis (feature extraction), parameter space and gain control (done via a set of rules);

Figure 3 is a schematic diagram showing an acoustic shock protection system with WOLA filterbank in accordance with a further embodiment of the present invention;

Figure 4 is a schematic diagram showing one example of the WOLA based architecture of Figure 3;

Figure 5 shows the block diagram structure of the sub-band adaptive periodic noise cancellation technique in accordance with an embodiment of the present invention; and

Figure 6 shows one example of the sub-band periodic noise cancellation block of Figure 5.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS:

An acoustic shock protection method, system and device in accordance with an embodiment of the present invention provides high fidelity and low group delay in the processed output signal relative to existing devices.

The acoustic shock protection method, system and device, in accordance with the embodiment of the present invention, protects a user against three basic forms of acoustic shock: 1- High-level, short duration (e.g. a loud sound); 2- Level versus time exposure; and 3- Rapid increase in relative level exposure.

Figure 1 shows an acoustic shock protection system 2 in accordance with an embodiment of the present invention. The acoustic shock protection system 2 includes an analysis block 4, a detection block 6, a removal block 8 and a logging block 10. The system 2 further includes a calibration block 12 that contains data used to calibrate the input and output levels and to support user preferences (for personalization). The arrows in Figure 1 show how these major blocks interconnect and interact with each other.

In the acoustic shock protection system 2, a novel pattern analysis based approach is employed to make the invention flexible and configurable and to provide high fidelity. The approach employs a collection of input signals (termed the "signal space"), which contains a representative subset of acoustic shock and non-acoustic shock signals that the system will process and, in the case of input signals that contain an acoustic shock event, protect against in typical operation. This signal space is processed via a feature extraction process (represented by the analysis block 4) to realize a reduced dataset representation of the signal space. The collection of reduced dataset representations (one for each of the signal in the signal space) is called the parameter space.

The analysis block 4 may use, for example, artificial neural networks, hidden markov models, Baysian estimation methods, signal statistics or combinations of signal parameters such as short-term and long-term level over various frequency ranges and time-scales. In some situations, it is necessary to measure and protect against level versus time exposure to signal levels. In combination with the other

subcomponents of the analysis block 4, a timer component 14 in the analysis block 4 provides the capability to protect against level-over-time exposure.

The detection block 6 contains a logical description in terms of inference rules representing patterns of analysis output conditions representing acoustic shock events. Detection block 6, for example, may take the form of a rule-based expert system.

The removal block 8 implements the signal processing necessary to remove the shock event from the input signal, typically by the application of time and frequency-dependent gain, while maintaining the fidelity of the output signal, relative to the input signal.

The logging block 10 implements on-line data collection of the key parameters of shock events. This information is time-stamped using the timer 14 or another similar means and may be categorized to provide accurate data records for later downloading and analysis.

The calibration block 12 provides data that is required to account for the acoustic and electrical differences between various headphones, headsets, systems and other components. The data from the block 12 ensures that the output level experienced by the user of the system 2 does not exceed a preset maximum as a function of time and frequency.

The logging block 10 and the calibration block 12 may be detachable.

In Figure 1, the analysis block 4, the detection block 6 and the removal block 8 are shown separately. However, the component of the analysis block 4 may be used as the component of the detection block 6 or the removal block 8. The component of the detection block 6 may be used as the component of the removal block 8. Processing done in each of the blocks may be reused in other blocks to minimize the computational and memory requirements for implementing the invention.

For example, the system 2 may re-use some of the features, which are used during analysis, as information (e.g. level measurements) to perform gain control in the removal block 8.

Measurements used to perform analysis and detection in the analysis block 4

and the detection block 6 are reused to perform the signal processing necessary for acoustic shock removal. Similarly, these measurements can be used as a portion of the data collected by the logging block 10.

The pattern analysis-based approach used for acoustic shock protection by the invention also provides the flexibility to effectively deal with a wide range of input signals types. This novel identification mechanism method effectively makes use of various signal parameters to identify acoustic shock events.

A feedforward configuration is used to provide fast-acting and intelligent gain control. In a preferred embodiment, the gain applied to the processed signal is controlled by a set of gain control rules that receive as input, the output(s) from the pattern analysis. While gain control is highly useful in removing acoustic shock when only a single signal is available, the addition of adaptive filtering provides further performance for periodic signal sources or in applications where a secondary input is available. The gain application and adaptive filtering blocks are seen as complementary. The gain control block acts quickly to remove acoustic shock transients. In cases where the shock is periodic noise or predictable from a secondary input, the adaptive filtering block converges to remove this noise. The details of the adaptive filtering block will be presented later, for now, the discussion will assume only a single input is available.

The entire implementation is parameterizable. This allows the flexibility need to deal with a wide range of environments and acoustic configurations.

Figure 2 shows the relationship between the signal space, analysis (feature extraction), parameter space and gain control (done via a set of rules).

The signal space 20 is mapped to the parameter space 22 defined by [X1, X2, X3, X4, X5..], by performing the feature extraction in the analysis block 4 of Figure 1. The detection block 6 makes a rule-based decision (rules 24) using the parameters. Then, by the removal block 8 of Figure 1, the gain control 26 is performed.

The use of this approach allows the system to be flexible and trainable. The signal space 20 can always be extended to account for a signal class or type for which protection must be provided. The rules 24 used for gain control 26 can be adjusted (in a preferred embodiment, they are done in software or firmware) to meet

new requirements.

An acoustic shock protection system using the aforementioned pattern analysis approach and implemented using a WOLA filterbank is now described in detail. The WOLA filterbank is described in United States Patent No. 6,236,731, "Filterbank Structure and Method for Filtering and Separating an Information Signal into Different Bands, Particularly for Audio Signal in Hearing Aids" by R. Brennan and T. Schneider, issued on May 22, 2001, and United States Patent No. 6,240,192, "Apparatus for and method of filtering in an digital hearing aid, including an application specific integrated circuit and a programmable digital signal processor" by R. Brennan and T. Schneider, issued on May 29, 2001.

The WOLA filterbank includes a WOLA analysis filterbank and a WOLA synthesis filterbank. The WOLA analysis filterbank receives a plurality of information signals in time domain and transforms the information signals into a plurality of channel signals in transform domain. The WOLA synthesis filterbank receives a plurality of channel signals in transform domain and transforms the channel signals into a single information signal in time domain.

Figure 3 shows an acoustic shock protection system using a WOLA filterbank in accordance with a further embodiment of the present invention. The acoustic shock protection system 40 shown in Figure 3 includes a feature extraction block 44, a shock detection block 46, a gain control logic block 48, a WOLA analysis filterbank 50, a WOLA synthesis 52 and a complex multiplier 54.

The feature extraction block 44 performs level measurements, fast and slow, broadband (in the time-domain) and narrowband (in the filterbank bands or groups of bands). The gain control logic block 48 selects appropriate level measurements and performs gain calculations.

The WOLA analysis filterbank 50 of the WOLA filterbank is used to divide the input signal into a plurality of frequency bands. These bands and the raw input signal are then processed individually or in combination to extract relevant signal features in the feature extraction block 44. This combination of processing steps implements the analysis block 4 shown in Figure 1. The results of this analysis are then passed to the shock detection block 46 which then makes a rule-based decision about the presence

or absence of an acoustic shock event. The shock detection block 46 thus performs the role of the detection block 6 in Figure 1. The shock detection decision is then used in conjunction with the extracted features to perform intelligent gain control. This is shown in Figure 3 as the gain control logic block 48, which is fed inputs by both the feature extraction 44 and the shock detection block 46. The application of the gain to these bands provides the control required to implement acoustic shock protection. The complexmultiplier 54 (used in real mode here) multiplies the output of the WOLA analysis filterbank 50 and the output of the gain control logic block 48. The WOLA synthesis block 52 of the WOLA filterbank receives the outputs of the multiplier 54 and transforms them into audio output in the time domain.

The use of the WOLA filterbank in this realization of the acoustic shock protection system provides high fidelity and low group delay. It also provides high adjacent band isolation that permits high-fidelity removal of frequency isolated shock events like tones. This implementation can be advantageously realized on the system architecture disclosed in U.S. Patent No. 6,236,731 and U.S. Patent No. 6,240,192.

Figure 4 shows one example of the WOLA based architecture shown in Figure 3. The acoustic shock protection system 40 of Figure 4 includes the feature extraction block 44, a shock detection and gain control logic block 80, and the WOLA filterbank which has the WOLA analysis filterbank 50, the multiplier 54 and the WOLA synthesis filterbank 52.

The feature extraction block 44 of Figure 4 includes fast and slow time-domain broadband level measurements and a vector of fast and slow narrowband level measurements corresponding to each frequency band.

The time domain measurements are now described in detail. In Figure 4, a first exponential average block ("Fast Exp. Avg.") 60, a slow exponential average block ("Slow Exp Avg.") 62 and a summer 64 implements the time domain measurements.

The time domain measurements can be first-order fast and slow exponential averages of the RMS (route mean square) signal level that are well-known in the art. Other level measurement techniques as known in the art are also possible.

The difference between these levels is a feature, which is sent to the shock decision block 74. Furthermore, these levels are calculated ahead of the WOLA processed signal using a delay block 72 ("z^{-n_0}" in Figure 4) that provides a time delay corresponding to n_0 samples. This delay permits the time-domain measurements to indicate the presence of a high level (and a potential acoustic shock event) to the shock detection rules before the shock reaches the filterbank. A fundamental property at the onset of any large acoustic transient or shock is that it is initially indistinguishable from a broadband disturbance. Only later, as the shock progresses, is further classification possible indicating the presence of narrowband/tonal noise or broadband noise. A rapid increase in time-domain energy, therefore, indicates that some form of shock condition is initiating and signals that a sensitization of the frequency domain detectors located in the frequency bands after the analysis stage should occur. This sensitization allows the gain logic to preemptively adapt eliminating a possible, brief high-level transient at the onset of a high level, because the gain control has already "seen" the high-level and had, if necessary, reduced the relevant gain or gains accordingly. The WOLA analysis filterbank 50 processes the output of the delay block 72. This parameterizable feed-forward delay allows the measurements to indicate the presence of shock one or more processing blocks before the shock would be (if unmodified) experienced by the end user, thereby improving protection. The inherent group delay of the WOLA analysis is used advantageously to reduce the delay time t implemented by the delay unit 72.

The frequency domain measurements are now described in detail. In Figure 4, a first exponential average block ("Fast Exp. Avg.") 66, a slow exponential average block ("Slow Exp Avg. ") 68 and a summer 70 implement the frequency domain measurements.

The frequency domain measurements can also be fast and slow exponential averages of the RMS signal level in each frequency band (as are well-known in the art). Other level measurement techniques are also possible. The difference between these levels is a feature, which is sent to the shock decision block 72.

The shock detection and gain control logic 80 of Figure 4 are now described in detail. The shock detection and gain control logic 80 includes the shock decision

block 74, a shock state machine 76 and a gain control block 78.

The shock decision block 74 makes a decision about the presence of acoustic shock at both the broadband and narrowband levels. A large value in either the time-domain feature or the band-level features indicates a quick increase in relative sound exposure level for the end user. In this realization, the thresholds are set by experimentation and heuristics. Alternatively, these could be set using automatic training methodologies such as artificial neural networks.

The shock decision is then used as part of a shock state machine 76, which uses a rule-base to decide whether the current exposure level is potentially harmful acoustic shock event. The shock state machine 76 coupled with the shock decision block 74 constitutes the detection block 6 shown in Figure 1.

The acoustic shock state, i.e. whether acoustic shock is currently occurring or not, is then used by the gain control logic (e.g. gain calculation 78 of Figure 4) to perform the activities of the removal block 8 of Figure 1. Output gain is reduced in bands where acoustic shock is detected in order to protect the user. The fast and slow level measurements from the feature extraction block, combined with instantaneous levels are used to compute the desired gain. Confining the output attenuation to only the bands where shock is prevalent minimizes any distortion of the output and thus provides improved fidelity.

This gain control details (such as maximum output level) and shock decision thresholds constitute the parameters that are available for personalization and calibration (e.g. block 12 of Figure 1) of the invention. The output of the shock decision can be used in the logging portion (e.g., block 10 of Figure 1) of the invention.

The detector (6 of Figure 1, 46 of Figure 3 and 74 of Figure 4) may use fuzzy logic, neural networks and/or combinations of overall-level and band-level control. Preferably, the frequency-domain processing is implemented with delay (72 of Figure 4) before the WOLA analysis filterbank 50 to obtain look-ahead functionality.

According to an aspect of the present invention, processing in a plurality of frequency domain bands provides the flexibility needed to effectively protect against wideband (in frequency) and narrowband shock signals. For narrowband signals (e.g.

tones or dual-tones), band-based gain control and/or adaptive processing provides localized gain control that results in high fidelity because gain adjustments are only made in the frequency ranges where they are necessary.

Periodic acoustic shock signals can be virtually eliminated using sub-band implementations of least-mean squares (LMS) periodic interference cancellation techniques (see, for example, B. Widrow and S. D. Stearns, "Adaptive signal processing." Englewood Cliffs, NJ: Prentice-Hall, 1985.). These can be implemented (block 8 in Figure 1) in subband outputs from the WOLA analysis as described below.

Figure 5 shows the block diagram structure of the sub-band adaptive periodic noise cancellation technique in accordance with an embodiment of the present invention. The audio signal contains the desired speech and acoustic shock event which is assumed to be a periodic noise source. Figure 5 is intended to illustrate the action of the periodic noise canceller. It is easily seen that this functionality may be added to the system of Figure 2 to obtain an improved acoustic shock cancellation system. As mentioned previously, the system of Figure 2 works quickly to remove the acoustical trauma while the addition of Figure 5 goes a step further to actually cancel the interference whenever a periodic acoustical shock is present.

A system shown in Figure 5 includes the WOLA analysis filterbank 50, the WOLA synthesis filterbank 52 and a processing block 92. The processing block 92 includes a plurality of sub-band periodic noise cancellation blocks 94A-94N, which corresponds to N sub-bands. The WOLA analysis filterbank 50 transforms audio input into a plurality of sub-bands. Each sub-band is then passed to the corresponding periodic noise cancellation block. The WOLA synthesis filterbank 52 combines the outputs of the processing block 92 into a single signal.

Figure 6 shows one example of the sub-band periodic noise cancellation block 94 of Figure 5. The sub-band periodic noise cancellation block 94 includes a delay block 96, an adaptive filter 98 and a block 100.

A delayed version of the audio sub-band input is filtered by the adaptive filter 98 and then subtracted from the original sub-band input by the block 100. The resulting error signal from the block 100 acts as the sub-band output and is also used to update the adaptive filter 98 using the LMS algorithm or other similar techniques.

The adaptive filter 98 will cancel the periodic noise because it is correlated with a time-delayed version of itself whereas the desired speech will remain because it is not similarly correlated with the delayed input.

In practice, it may be desirable to only use the filtered output of summer junction 100 when strong periodicity is detected. When weak or no periodicity is detected, the sub-band output should be made equal to the sub-band input to avoid degradation of the signal. This periodicity detector could be made by allowing the sub-band filtering block to always operate while observing the variance of the filter coefficients. During regions where the variance is high, above a settable threshold, periodicity would be declared, activating a switch to the filtered output. Conversely, where the variance is low, below a settable threshold which may be different from the previous threshold to implement hysteresis, periodicity would be declared absent and switching the output to use the unfiltered input. Other versions are envisaged as well, where the periodicity detection is done with fuzzy logic and the output is a weighted sum of the filtered and unfiltered inputs according to the certainty of the periodicity detection.

According to a further aspect of the present invention, the use of a patented oversampled filterbank (US patent Nos. 6,236,731 and 6,240,192) also provides low-group delay and the ability to make large gain adjustments without adversely impacting signal fidelity or adjacent band signals. These combine to offer a high level of protection and high fidelity.

Further, calibration and personalization allow the invention to be easily adjusted to work with a specific acoustic configuration (eg, a specific headset or headphone). Built-in calibration software allows for simplified calibration in the field. Personalization allows some parameters of the invention to be adjusted by the user so that the performance and fidelity suit their specific preferences.

Finally, the embodiment of the present invention provides the ability to log data and download it for later analysis. This can be done over a wired or wireless (radio frequency or other) link. For example, the acoustic shock protection system can be connected to an IP network (and to the Internet) or it can use existing RF technologies, such as 802.11, Bluetooth or Zigbee. An addressing scheme allows

the logging and analysis of data from multiple systems so that the invention can be used in an environment where a number of units need to have data logged and analyzed (e.g. a call center).

The acoustic shock protection system of the present invention is applicable to telephone systems (both mobile and land), mobile and land-based radio systems, audio protection systems and similar devices, such as headsets or headphones, that protect users from loud sounds. Similar areas of technology include audio limiters and dynamic range compressors.

The detail of the present invention is further understood with reference to the attached Appendixes A and B.

The acoustic shock protection system (device) of the present invention may be implemented by any hardware, software or a combination of hardware and software having the above described functions. The software code, either in its entirety or a part thereof, may be stored in a computer readable memory. Further, a computer data signal representing the software code which may be embedded in a carrier wave may be transmitted via a communication network. Such a computer readable memory and a computer data signal are also within the scope of the present invention, as well as the hardware, software and the combination thereof.

While particular embodiments of the present invention have been shown and described, changes and modifications may be made to such embodiments without departing from the true scope of the invention.

What is claimed is:

1. A method of providing a protection against acoustic shock, the method comprising the steps of:

performing a pattern analysis to an input signal to identify a parameter space corresponding to a signal space of the input signal;

applying a rule-based decision to the parameter space to detect an acoustic shock event; and

removing the acoustic shock event.

2. A method of claim 1 wherein the step of performing a pattern analysis includes the step of performing a feature extraction from the input signal to identify the parameter space.

3. A method of claim 2 wherein the step of removing the acoustic shock event includes the step of performing gain control.

5. A method of claim 1 further comprising the step of performing calibration to keep an output signal provided to a user at a specific level.

6. A method of claim 1 further comprising the step of implementing on-line data collection of the acoustic shock events from the input signal.

7. A method of providing a protection against acoustic shock, the method comprising the steps of:

performing a weighted overlap-add (WOLA) analysis on an input signal;

performing a feature extraction to the input signal and the band signals provided by the WOLA analysis;

detecting an acoustic shock event based on the feature extraction;

performing gain control based on the feature extraction and the shock detection;

applying a calibrated gain to the signal to reduce the level to meet a predetermined safe level; and

performing a WOLA synthesis to the band signal and a signal provided by the gain control to synthesize an output signal.

8. A method of claim 7 wherein the step of detecting an acoustic shock event detects an acoustic shock event based on the feature extraction using a rule-based decision.

9. A method of claim 7 further comprising the step of delaying the input signal to the WOLA analysis to allow time to obtain fast broadband features to aid in the interpretation of the WOLA analysis results.

10. A device for providing a protection against acoustic shock, the device comprising:

an analysis module for performing a pattern analysis on an input signal to identify a parameter space from a signal space of the input signal;

a detection module for applying a rule-based decision to the parameter space to detect an acoustic shock event; and

a removal module for removing the acoustic shock event.

11. The device of claim 10 wherein the analysis module performs a feature extraction from the input signal to identify the parameter space.

12. The device of claim 10 wherein the detection module performs gain control using a state machine that detects acoustic shock events.

13. The device of claim 10 further comprising a removable module for removing the acoustic shock event.

14. The device of claim 10 further comprising a calibration module for performing calibration to keep an output signal provided to a user at a specific level.

15. The device of claim 10 further comprising a logging module for implementing on-line data collection of the acoustic shock events from the input signal.

16. The device of claim 10 further comprising a module for performing weighted overlap analysis and synthesis to implement processing in sub-bands.

17. A device for providing a protection against acoustic shock, the device comprising:

a weighted overlap add (WOLA) analysis module for transforming the input signal to a band signal;

a feature extraction module for performing a feature extraction to an input signal and the band signal;

a detection module for detecting an acoustic shock event based on the feature extraction;

a gain control module for performing gain control based on the feature extraction and the shock detection; and

a gain control unit for applying a calibrated gain to the signal to reduce the level to meet a predetermined safe level; and

a WOLA synthesis module for synthesizing the band signals to provide an output signal.

18. The device of claim 17 wherein the detecting module detects an acoustic shock event based on the feature extraction using a rule-based decision.

19. A device of claim 17 further comprising the step of delaying the input signal to the WOLA analysis to allow time to obtain fast broadband features to aid in the interpretation of the WOLA analysis results.

20. A method of providing a protection against an acoustic shock, the method comprising the steps of:

transforming an input signal into a plurality of oversampled sub-band signals in a frequency domain;

adaptively processing the sub-band signals to remove an acoustic shock event:

combining the processed sub-band signals to generate the output signal.

21. A method of claim 20 wherein the step of processing the sub-band signals includes the step of processing each sub-band signal to remove a periodic acoustic shock event.

22. A method of claim 21 wherein the step of processing the sub-band signals includes the step of delaying the sub-band signal, the step of adaptively filtering the delayed sub-band signal in a filter and the step of adding the sub-band signal and the output of the filter.

23. A method of claim 22 further comprising the step of adjusting the filter.

24. A device for providing a protection against an acoustic shock, the device comprising:

a weighted overlap add (WOLA) analysis module for transforming an input signal into a plurality of oversampled sub-band signals in a frequency domain;

a processing module for adaptively processing the sub-band signals to remove an acoustic shock event:

a WOLA synthesis module for synthesizing the processed sub-band signals to provide an output signal.

25. The device of claim 24 wherein the processing module includes a plurality of sub-band periodic acoustic shock cancellation module, each of which processing the

corresponds sub-band signal.

26. The device of claim 25 wherein the sub-band periodic acoustic shock cancellation module includes a delay module for delaying the sub-band signal, an adaptive filter for adaptively filtering the delayed sub-band signal and a summer for adding the sub-band signal and the output of the filter.

27. The device of claim 26 further comprising a module for adjusting the filter.

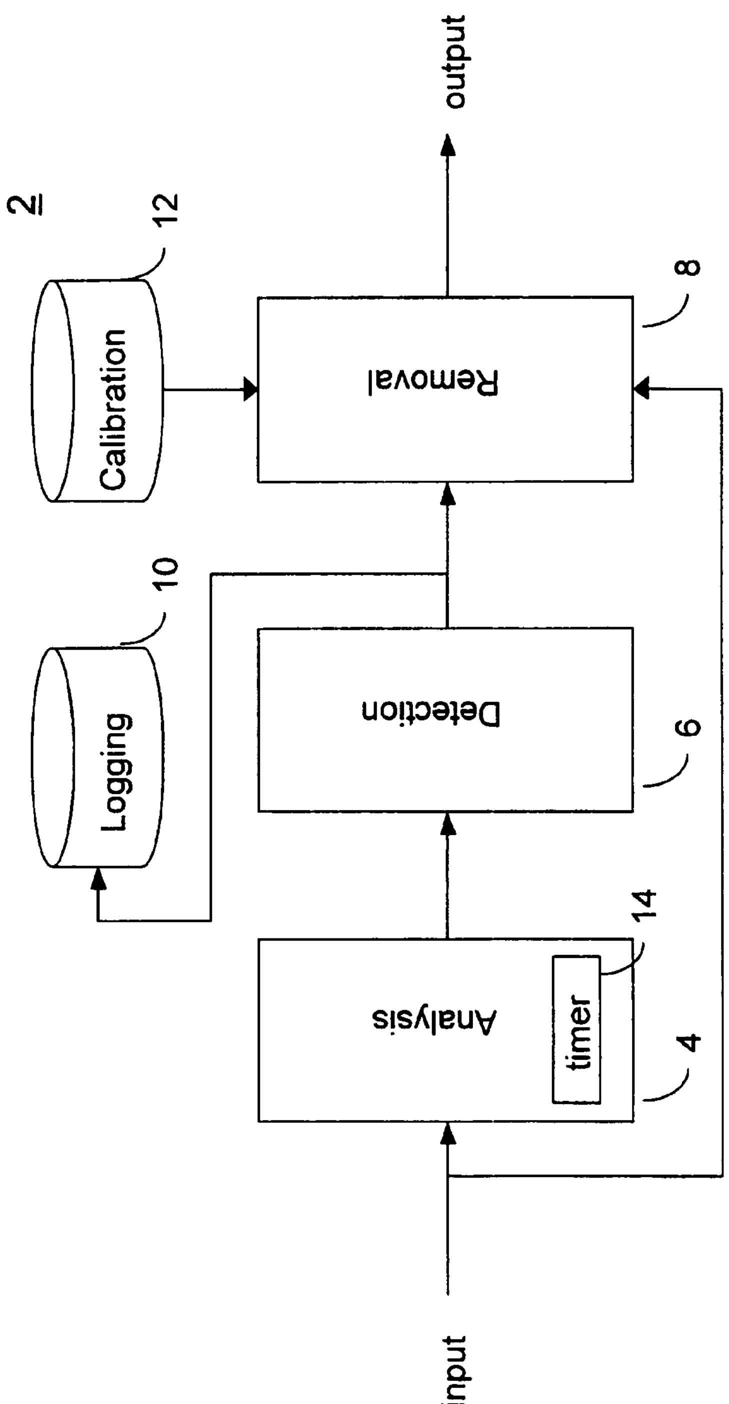
2
Calibration
12
Logging
10
Removal
8
Detection
6
Analysis
timer
14
4
input
output

FIGURE 1

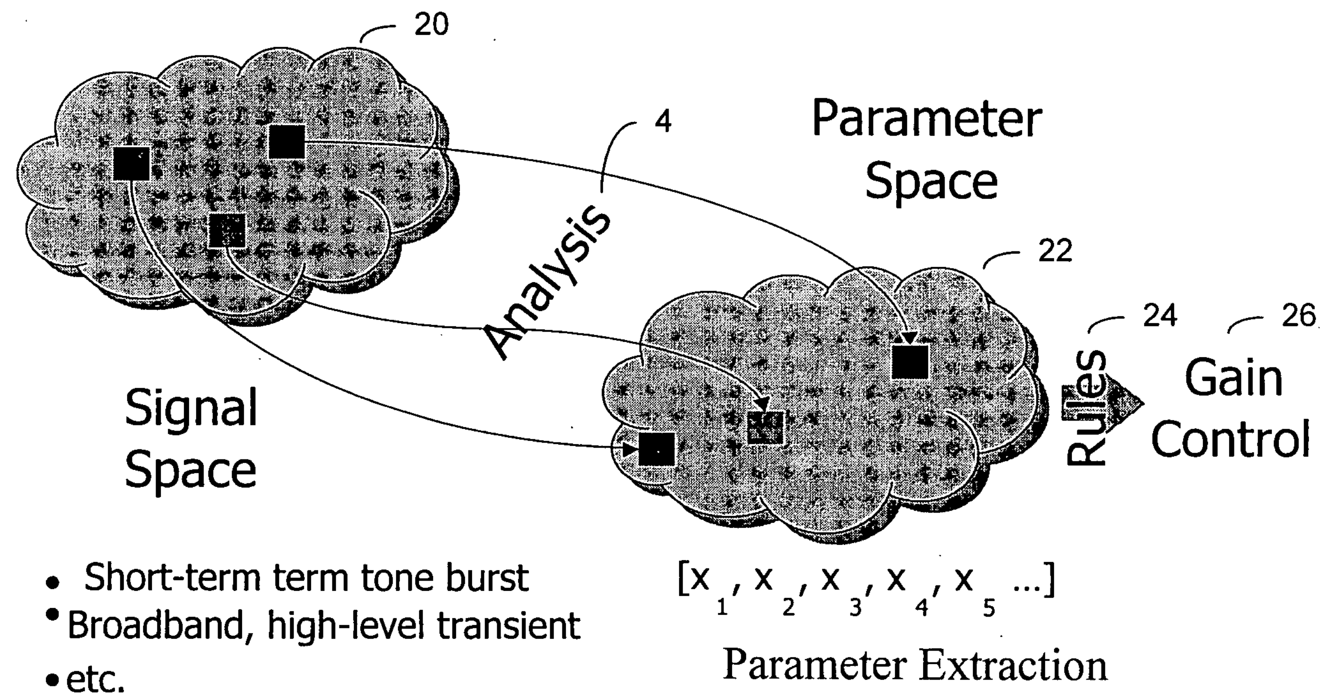
20
4
Parameter
Space
22
24
26
Analysis
Rules
Gain
Control
Signal
Space
• Short-term term tone burst
• Broadband, high-level transient
• etc.
[x1, x2, x3, x4, x5 ...]
Parameter Extraction

FIGURE 2

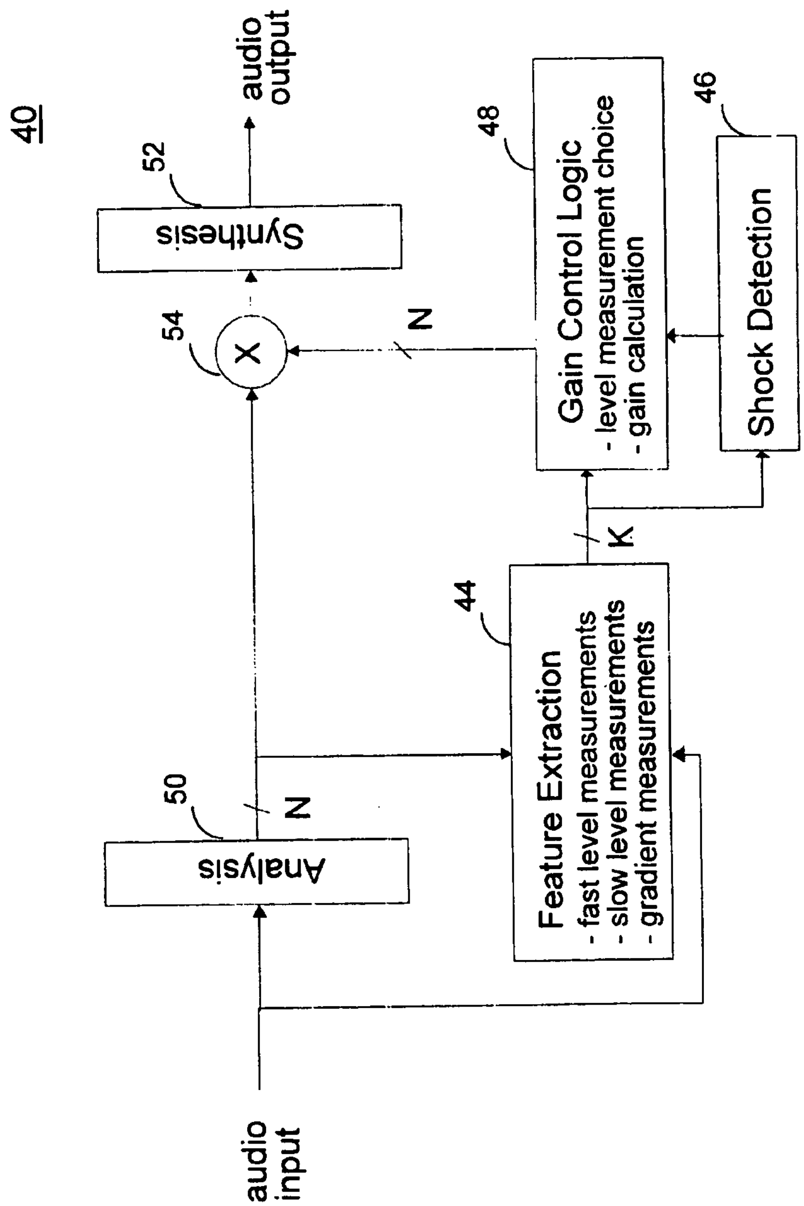
40
audio input
Analysis
50
N
Feature Extraction
- fast level measurements
- slow level measurements
- gradient measurements
44
K
Gain Control Logic
- level measurement choice
- gain calculation
48
Shock Detection
46
N
X
54
Synthesis
52
audio output

FIGURE 3

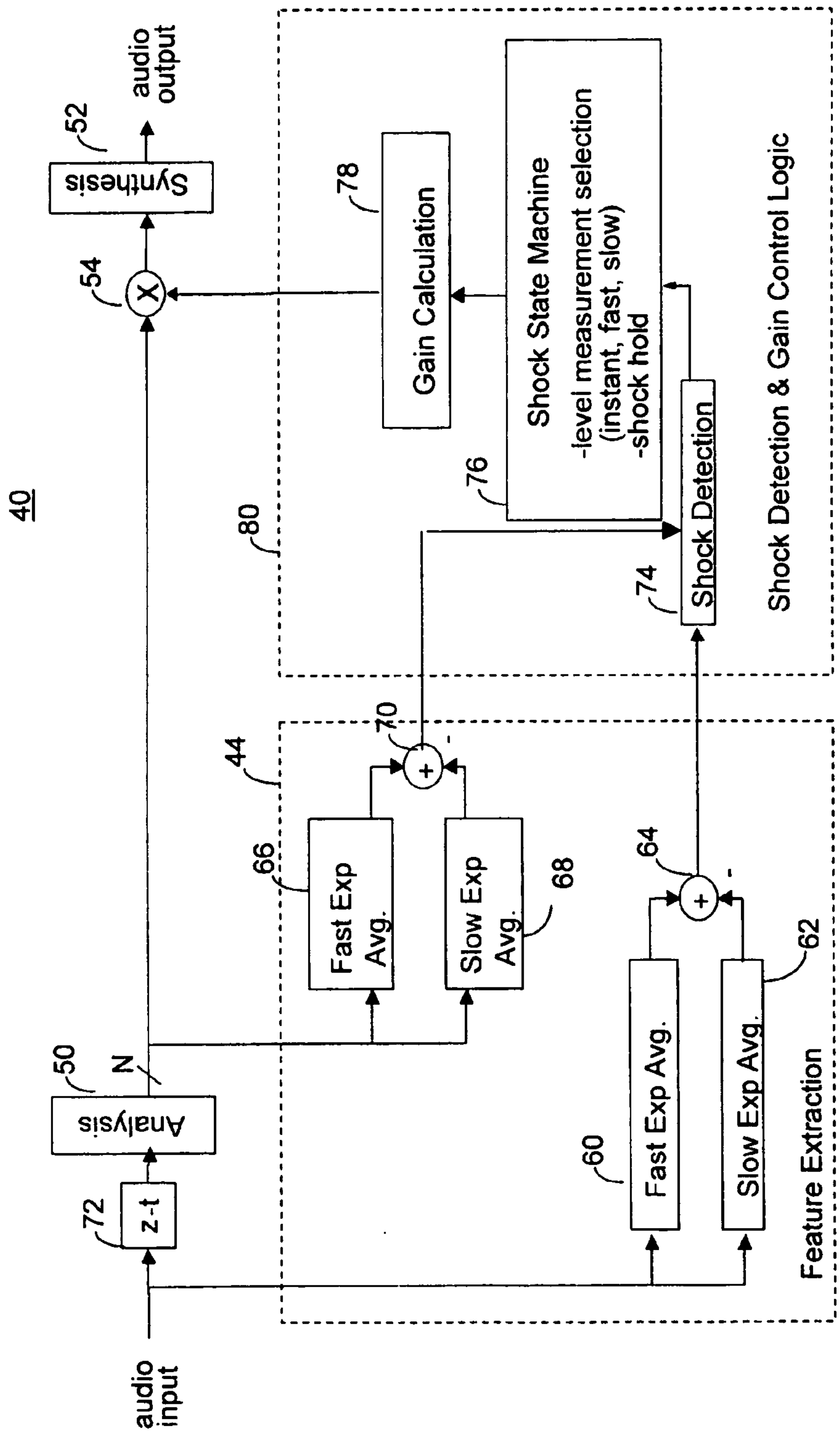
40
audio input
72
z-t
50
Analysis
N
52
Synthesis
audio output
54
X
44
66
Fast Exp Avg.
68
Slow Exp Avg.
70
+
-
60
Fast Exp Avg.
62
Slow Exp Avg.
64
+
-
Feature Extraction
80
78
Gain Calculation
76
Shock State Machine
-level measurement selection
(instant, fast, slow)
-shock hold
74
Shock Detection
Shock Detection & Gain Control Logic

FIGURE 4

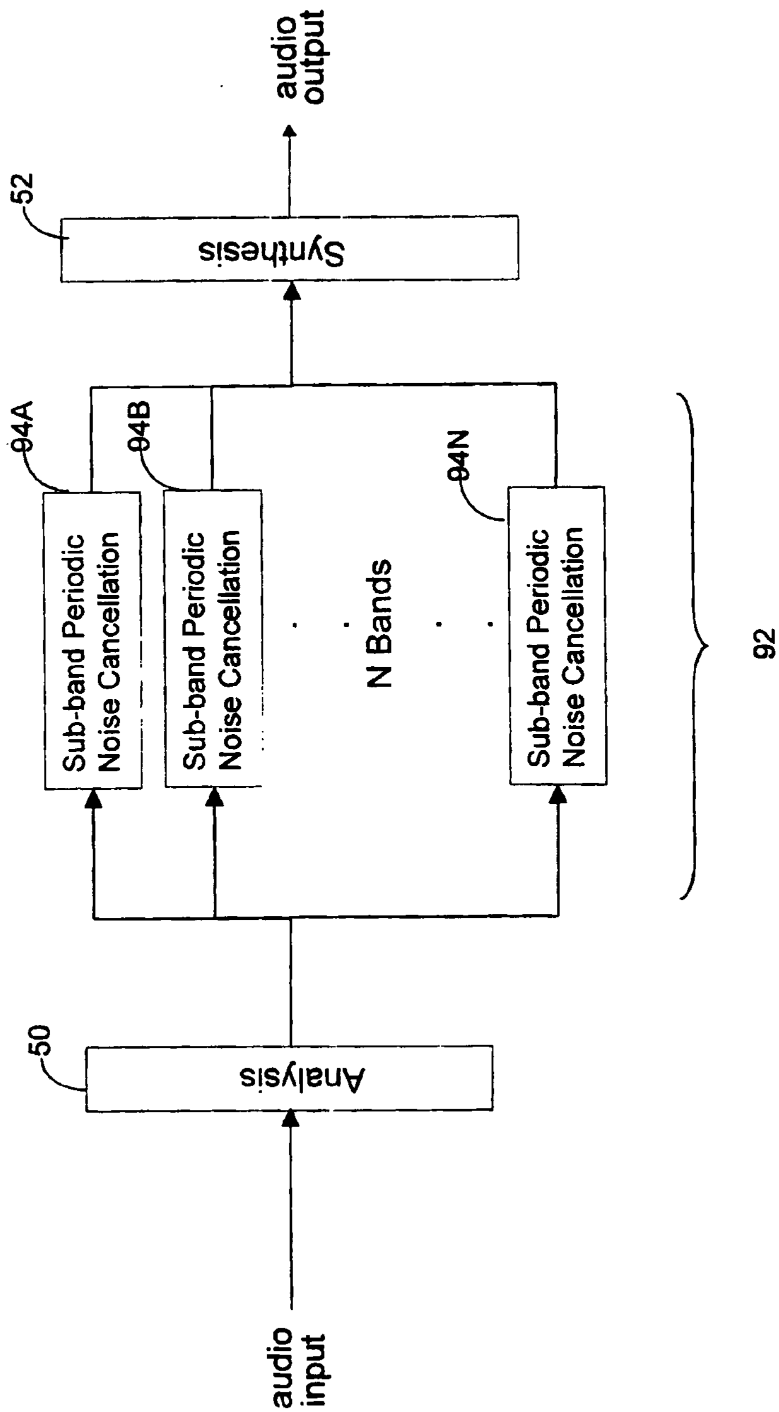
audio input
Analysis
50
Sub-band Periodic Noise Cancellation
94A
Sub-band Periodic Noise Cancellation
94B
N Bands
Sub-band Periodic Noise Cancellation
94N
92
Synthesis
52
audio output

FIGURE 5

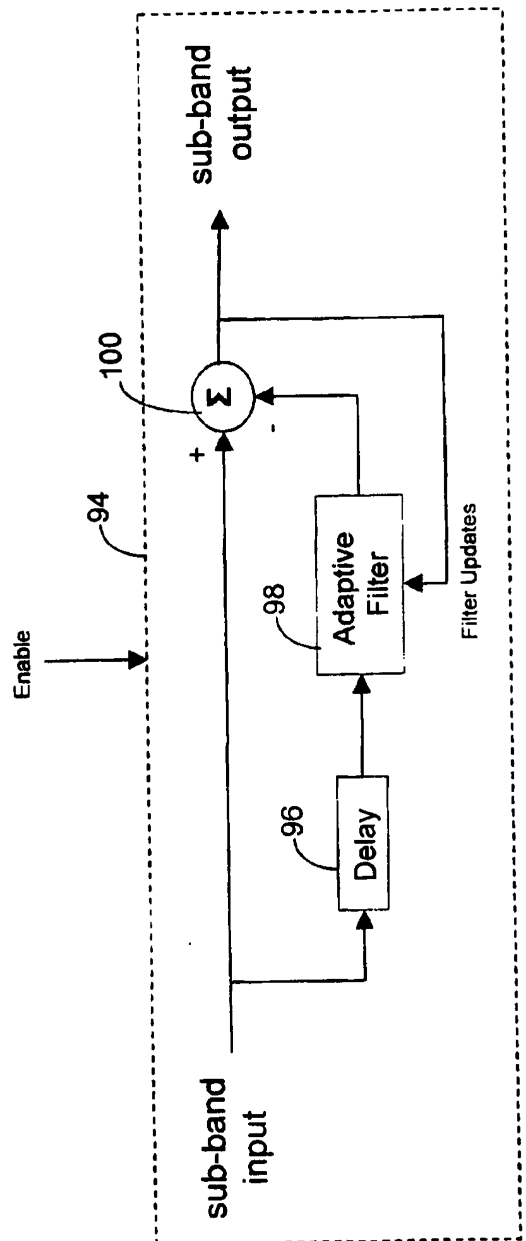
Enable
94
sub-band input
96
Delay
98
Adaptive Filter
Filter Updates
100
Σ
+
-
sub-band output

FIGURE 6